A brief introduction to audio systems

Gary Tuttle

gtuttle@iastate.edu

https://gtuttle.net/audio

What are the parts of a basic audio system?

- Stored source material
- Digital to Analog conversion
- Transmission to an amplifier
- Amplification
- Electrical to acoustical transducer (a speaker)
- Transmission through the air to the listener's ear

Design considerations

- Power / efficiency
- Distortion
- Frequency response & signal processing

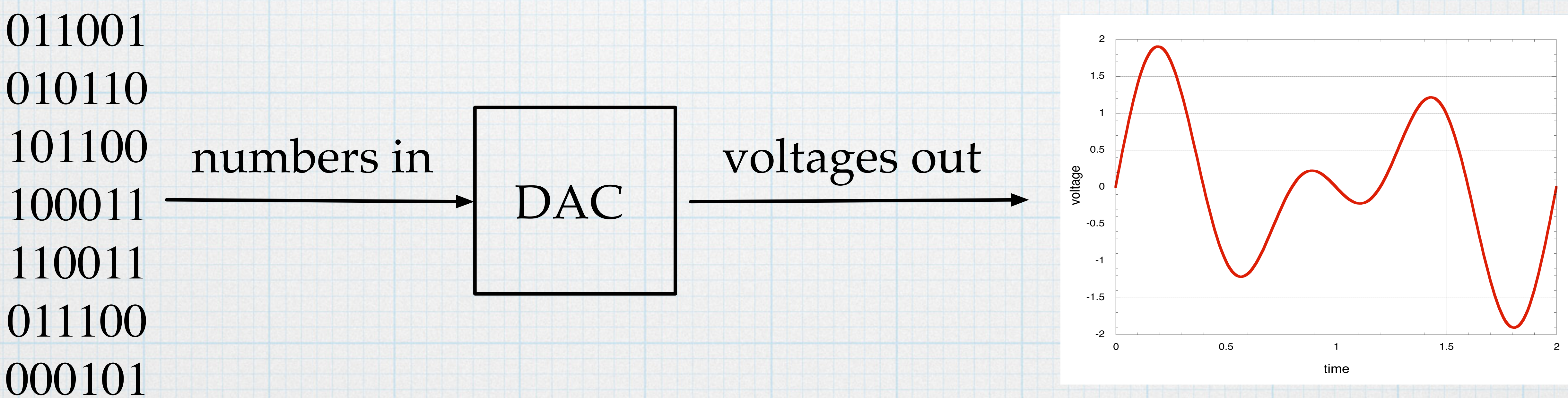
Some important considerations

- Our ears are analog — they respond to a continuous series of pressure vibrations arriving from the surrounding air.
- Most modern audio storage is digital — numbers stored in some form of electronic memory element. Needs to be converted to analog.
- Humans can hear vibrations that are oscillating at frequencies between 20 Hz and 20,000 Hz. Frequencies below are “infra-sonic” (elephants) and those above are “ultra-sonic” (dogs). All aspects of an audio system should work properly for all frequencies in this range. (Note, though, that the human range of hearing changes with age.)
- Because we have two ears, we are able to “image” sound and determine the direction that it is coming from. (At least for some frequencies.)
- The range of “loudness” (acoustic power) that we can detect and tolerate is huge. If we assign the smallest possible sound that a typical human can hear as 1, then the loudest that we can handle is 10^{12} (one trillion). (Above this, our ears may be damaged.) Using technical jargon, the range is 120 dB.

Audio storage

Digital-to-analog conversion

- Necessary to go from modern data storage to ancient analog ears.
- Can happen at different points along the audio system path, but must happen somewhere.
- Requires an electronic circuit — digit-to-analog converter (DAC). (EE 230)
- Converting in an effective manner requires some signal processing. (EE 224 and up).



- Often the digital data is stored in a “lossy" format (mp3, aac, or similar) and must be converted back to a native format before using the DAC. (Digital signal processing, EE 424).

Transmission to the amplifier

- Traditionally a wire was used, and still very common. Still probably the best way to transmit the signal with no distortion or data loss.
- However, an alternative method has become available in recent years — wireless transmission using Bluetooth, Wifi, or some other proprietary communication scheme (eg Sonos). With modern wireless systems, the data is transmitted in digital form, so the D-to-A conversion takes place *after* transmission.
- Requires all the elements of a standard communication system — transmitter and receiver circuits and antennas at both ends.
- There are analog transmission systems still in use — AM and FM radio.
- There are also optical fiber transmission systems which send the info via light pulses. These also require transmitters and receivers.
- Wireless transmission often requires some sort of data compression, so there is possibility of information loss.

The amplifier

- The signals coming from the source are too weak to drive typical speakers. Making a "normal" desktop or living-room speaker give out some sound requires at least a few tenths of a watt of power. The audio signal from a DAC might have only a few milliwatts of power available. The amplifier adds power to the signal. (Remember: $P = V \cdot I$, so both voltage and current must be boosted.)
- The amp takes energy from a DC power supply and adds it to the audio signal. The DC power can come from a battery or from an AC-to-DC converter that is plugged into a wall outlet.
- Consideration: total power available. More power requires bigger power supplies and bigger components. This adds cost and size.
- Consideration: efficiency. If running from a battery, efficiency is crucial. Inefficient amps may require significant heat sinking to protect from getting too hot. This also adds to cost and size. (class A vs. class B vs. class D.)
- Consideration: distortion. Amplifiers are inherently non-linear, meaning that the signal might be changed as it passes through the amplifier. Usually, this is bad. It is expressed in terms of total harmonic distortion. (THD)
- Consideration: frequency response. The amp should amplify all frequencies the same amount.

The speaker (acoustic transducer)

- Once the audio signal has enough power (voltage and current), it can used to drive a speaker.
- The speaker is essentially a linear motor, converting electrical energy into mechanical energy and then into acoustic energy. https://www.youtube.com/watch?v=AP2Nu4MZJRs
- Speaker design requires consideration of several inter-related topics: filtering of the audio signal, magnetics, mechanical vibration, and interference of sound waves. For something that is so common and outwardly simple, the detailed operation is quite complex.
- Consideration: total power. The coil, magnet, and speaker cone must sized to match the desired acoustic power output — more power = bigger (and more expensive) components.
- Consideration: distortion. Like with the amplifier, the speaker should not change the signal as it is changing from one form to another.
- Consideration: multiple drivers. Because a single driver cannot produce all frequencies (due to the limits of mechanical vibrations), a speaker may have multiple drivers of different sizes. Each size will produce a particular frequency range. In this case, the electrical signal has to be modified so that only the appropriate frequencies are passed to each driver.

Sound waves

- The audio signal travels as energy through the air as sounds waves — moving pressure variations.
- All waves have a characteristic wavelength that is related to the frequency and speed that wave at which the wave travels, $\lambda = v / f$. (v = 343 m/s = 767 mph). At f = 20 Hz, λ = 17 m = 56 ft. At f = 20 kHz, λ = 1.7 cm = 0.67 inches.
- The three-orders-of-magnitude difference in the wavelengths has important implications for how we perceive the sounds. The short wavelengths of the high-frequency sounds mean that they can be viewed as traveling almost like rays. (We often treat light waves as being ray like.) This means that we can discern the direction the high frequency sounds comes from. The long wavelengths of low frequencies make determining directions impossible.
- Speakers are most effective when their physical size is commensurate with the wavelengths they are producing. While a 56-ft diameter speaker is not practical — although it would be awesome! — the general trends hold. Small diameter speakers are used to produce higher-frquency sounds and bigger speakers produce lower sounds. A three-way system is common, a small "tweeter", a medium-sized "mid-range", and a big "woofer" or "sub-woofer".
- The wave nature of sound also means that reflections will lead to constructive and destructive interference. So the shape of a room — and hence the shape of the reflections — can have big impact on what we hear.

Stereophonic sound

- The fact that we have two ears separated by the width of our head means that sounds reaches the two “sensors”at slightly different times. We can use the time difference to determine the direction that the sounds comes from. At least for higher frequencies.
- When listening to a live performance, we can discern where the drums are, where the strings are located, and where the horn section is seated on the stage.
- We can use two speakers (stereo) when playing recorded music to “fake” the effect of listening to a live performance. It is easy to separate the various sounds from a live performance and record those into two separate “channels” — some sounds go into the “right” channel and some into the “left”. If the two channels are played back through two speakers that are spaced apart in front of us, we will hear slightly different sounds coming from each speaker. Our brain will combine the sounds in such a way that we perceive a spread-out orchestra in front. It is an illusion, but a very potent one. Almost all music is recorded in “stereo” for this reason. (If the sounds are not separated, then the audio is said to “mono”.
- The effect can be extended to more speakers — a quadraphonic system or surround sound that comes with video. But stereo is the most common for straight audio applications.

To review:

For any audio system, we need:

- Stored stereo audio source
- Digital to Analog conversion (probably)
- Transmission to an amplifier (wire or bluetooth)
- Amplifier
- Speakers
- Transmission through the air to the listener's ear

Each of these things will affect the quality of the sound we hear. The important factors are distortion and frequency response.

Which is most important?